SPECIFICATION

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VOICE OVER IP TELEPHONE RECORDING ARCHITECTURE

Background of Invention

[0001] The present invention relates to a distributed network architecture for recording a stream of analog or digital data.

[0002] In the current state of the industry, in order to record the contents of a telephone call, computer transmission, television program, or any form or type of analog or digital data transmission, there are two basic methods. The first method is "logging," which consists of recording every transmission, from initiation until termination, regardless of the content or characteristics of the transmission. Because logging is independent of the contents and characteristics of the transmission, if a specific transmission is to be reviewed, it becomes necessary to record every transmission. Recording every transmission, however, requires a significant amount of storage space. Further, because of the number of transmissions recorded, it is difficult and time-consuming to locate a specific transmission.

[0003] The second method of recording a transmission is "event driven" recording, which records a transmission upon the occurrence of a certain condition or event. In some instances, the recording condition may be the start of the transmission (which is, in effect, the logging method), while in other instances the event will occur during the transmission (such as when a user presses a "record" button during the transmission). Other events may be time-based, such as when recording begins and ends at a specific times. One difficulty of an event driven recording system is properly defining the event that initiates recording. An overly broad event may record too many transmissions, while too specific of an event may not record enough. Another disadvantage of event driven recording is that the event may not occur, or may occur

at the wrong time, resulting in not all of the content of the transmission being recorded. Yet another disadvantage is that if the event cannot (or does not) occur until after the initiation of transmission, then the transmission that occurs prior to the event is not recorded. Further, for communications via computer instant messenging, a similar disadvantage occurs when the content of the conversation may "scroll off" the screen before the user can save the earlier portions of the conversation.

[0004] The above described recording methods all deal with a centralized network architecture. That is, the functional components that manage call control and voice transmission are all centrally located in a Private Branch Exchange (PBX) of a private network. More specifically, a recording initiation device such as a recording phone has to obtain all call control information and voice data for other phones in connection therewith from the PBX.

[0005] What is needed is distributed network architecture for recording voice data between telephones.

Summary of Invention

[0006] A method and system is disclosed for on-demand recording of a voice session by a telephone recording device in a telecommunication network. After establishing a voice session between the telephone recording device and at least one communication device, a user of the telephone recording device may instruct it to store voice data during the voice session so long as the voice session has not been terminated. During the voice session, the telephone recording device processes and transmits the voice data to and saved at a storage server without going through a centrally located exchange device, wherein the saved voice data is available for on-demand replay.

Brief Description of Drawings

[0007] Fig. 1 illustrates a conventional telephone recording system using a centrally located exchange device.

[0008] Fig. 2 illustrates a distributed recording system in a telecommunication network using a recording phone.

[0009] Fig. 3 illustrates a flow diagram for implementing the on-demand recording of a

voice session according to one embodiment of the present invention.

[0010] Fig. 4 illustrates a distributed recording system in a telecommunication network using a recording phone according to another embodiment of the present invention.

[0011] Fig. 5 illustrates a distributed recording system in a telecommunication network using a recording phone according to another embodiment of the present invention.

Detailed Description

[0012] The following disclosure provides many different embodiments, or examples, for implementing different features of an on-demand recording system. Specific examples of components, processes, and implementations are described to help clarify the invention. These are, of course, merely examples and are not intended to limit the invention from that described in the claims.

[0013] Referring to Fig. 1, a centralized on-demand recording system 100 includes an centrally located exchange device 10 connected to a server 12 via a network 14. The centrally located exchange device 10 such as a PBX includes a memory buffer 16, such as RAM or a hard drive, a processor 18 for converting analog signals into digital signals, and for placing data into packets. While only one processor 18 is shown, it is contemplated that a processing system, composed of one or more processors in conjunction with firmware or software, could provide equivalent functionality. The server 12 includes a memory buffer 20, such as random access memory (RAM) or a hard drive, a storage device 22, such as a compact disk (CD-ROM) drive, floppy drive, or hard drive, and a device 24 for receiving a save initiation request. The centrally located exchange device 10 is connected to one or more communication devices 28 via the network 14. Communication devices 28 may be capable of transmitting or receiving analog or digital signals via the network 14. The network may be any type of wired or wireless system for transmission of signals, including, a plain old telephone switch (POTS) network, computer packet network, or television broadcast system.

[0014] Connected to at least one of the communication devices 28 is a save initiator 30, which allows a user to initiate a save request. While depicted in Fig. 1 as being connected to one of the communication devices 28, the save initiator 30 may instead (or in addition) be connected to any of the other components or may be a stand-alone

component connected to the network, thus allowing the initiation of the save request by someone other than a user of the communication devices 28.

[0015] While two parties are engaging in a voice session, the corresponding voice data is duplicated and digitized by the processor 18 and may be placed into packets. The packets are then provided over the network and received by the server 12. The server 20 extracts (e.g., de-packetizes) the digitized information from the packets and stores the extracted information in the memory buffer 20. If a save request signal is initiated, the save initiator 30 transmits a save request signal for receipt by the receiver 28. When the connection is terminated, the contents of the memory buffer 20 on the server 20 is copied onto the storage device 22. The saved information of the entire connection on storage device 22 is made available for playback via devices such as the communication device 28.

[0016] In an enterprise environment wherein a complex telephone system is deployed, any number of telephones 28 may be connected to any number of line cards of the centrally located exchange device 10. The line cards may contain functional modules that deliver that functions equivalent to the memory module 16, digitizing processor and packetizing processor system 18.

[0017]

Fig. 2 is a graphic representation for a distributed Voice Over IP (VoIP) telephone network 200 including a telephone recording device such as a VoIP recording phone 202 and other communication devices such as regular VoIP telephones 204a-d, which may not have the recording feature. Although only one VoIP recording phone 202 is shown, it is understood that a number of such phones can be connected to the network 200. It is assumed that telephones 204 a-c are connected to a hub device 206a, and telephone 204d and the recording telephone 202 are connected to another hub device 206b. The hub device 206a or 206b can be an Ethernet hub device that are normally used to connect various computing devices such as another hub device, a storage server such as an IDVR server 208, or a call control unit such as a call manager 210. The VoIP recording telephone 202 performs an on-demand recording process to process and send all voice data from any other telephone involved in a conversation therewith to the IDVR server for recording purposes without going through a centrally located exchange device (such as the server 12 of Fig. 1). The

recording telephone 202 also obtains call control information from the call manager 210 without involving a centrally located exchange device.

[0018] When a voice session is established between two telephones transmitting voice data, while at least one of which is a VoIP recording telephone 202 and the other is a regular phone such as telephone 204a, the voice data initiated by telephone 204a goes to a local switch device such as the hub 206a, further through another hub 206b, and reaches the recording phone 202. The recording phone 202 will dynamically process and duplicate the voice data received. If the voice data is not already in digital form, it is digitized and packetized by the recording phone 202, and sent to the IDVR server 208. The IDVR server 208 will extract (e.g., de-packetize) the digitized voice data from the packets and store the data in a memory buffering device such as a memory module in the IDVR server. If needed, the voice data is further stored in a persistent memory device for future replay.

[0019] Fig. 3 illustrates a flow diagram 300 for recording a voice session over a VoIP recording network 200. After a voice session is established in step 302, the voice data (either initiated by the recording phone or other noon-recording phone) is packetized by the VoIP recording phone 202 and sent to the IDVR server for temporary storage. A temporary memory buffering device such as a memory buffer of the IDVR server may be used to tentatively hold the stored information. At any time during the voice session and up until the session is terminated, the user of the recording phone 202 may request that the information of the entire voice session be saved for later retrieval by initiating a save request in step 304. To initiate the save request, the user may only need to push a predetermined button on the recording phone 202. Other user interfaces may also be possible to send the save request. Although the VoIP recording phone is the instrument processes the save request, it is noted that it doesn"t have to be the user of the VoIP recording phone who initiates the request, a user on the other end of the line using a regular VoIP telephone can also initiate the process by sending a notice to the VoIP recording phone. Since the packets are used as the transport means for information exchange, it should be understood by one skilled in the art that all other VoIP telephones can send a signal to the VoIP recording phone to request a conversation session to be saved. It should be noted, however, that the user of the VoIP telephone recording system does not necessarily have to be a user of a telephone

204a-d or 202. The user can use a computer terminal that can communicate with the VoIP recording phone. In step 306, the save request signal is sent to and received by the IDVR server. Upon receiving the instruction, the IDVR server stores all the voice data currently stored in the temporary memory module and any forthcoming voice data in a predetermined persistent storage device of the IDVR server for future retrieval (step 308). On the other hand, if the save request is never initiated before the end of the voice session, the voice data tentatively stored by the IDVR server will be deleted when the voice session ends. The user can retrieve the stored voice data at any time from the IDVR server through any compatible playback device such as a regular VoIP telephone (step 310). In this manner, regardless of when the save request is initiated during the session, the entire communication of the voice session is available for playback. It is also understood that the temporary memory buffering device does not need to be on the IDVR server, it can reside in the VoIP phone as well. If so, the VoIP recording phone processes the voice data and keeps it until it is clear whether the voice data needs to be sent to the IDVR server for storage. In other words, if the voice session ends without a save request ever issued, the VoIP phone discards the content tentatively saved thereon. The hubs shown can be replaced by switches if required, and the voice session can still be recorded based on the above described principles. Fig. 3 thus illustrates a distributed communication architecture (e.g., most likely within a company) wherein two or more people can start and record a peer-to-peer conversation without involving a centrally located exchange device such as a private branch exchange (PBX) or any similar devices.

[0020] Fig. 4 illustrates a peer-to-peer VoIP telephone recording system 400 involving only one local switch device. The system 400 is similar to the system 200 of Fig. 2 except that only one switch 402 is involved. The switches, like the hubs, operate at the Ethernet level for the communication, and only serve the purpose of routing the packets from one end to the other of an established communication link. In this configuration, a voice session is carried out with voice data transmitted only through one single switch. No centrally located exchange device is involved at all.

[0021] Fig. 5 illustrates a VoIP telephone recording system 500 according to another embodiment of the present invention. This system 500 connects a VoIP recording phone with a remote VoIP phone through a series of network devices such as the

switches 402a-b, a gateway device 502, and an IP transport network 504. This illustrates that the voice recording mechanism as described with Fig. 3 can be extended to a much bigger network or even to multiple networks as long as one VoIP recording phone is operating together with an IDVR server 208 and a call manager 210, which manages the voice session. It is also understood that since the VoIP recording phone transmits packets to the IDVR server 208, the IDVR server does not have to be physically located in the same network or in the vicinity of the recording phone. As the packets can travel a long distance in a short time period, the IDVR server can actually be located in a different network.

[0022] The VoIP recording phone 202 shown in Figs. 2-5 can be easily replaced by computing devices such as personal computers or personal digital assistants as long as such computer device has the VoIP recording telephone function. While packets are used in the preferred embodiment above, other forms of data transfer are commonly used by those skilled in the art, including, for example, frames, raw data, or tokens.

[0023] The present invention as described above thus provides an improved method for allowing the entire telephone conversation, and other streams of analog or digital data to be recorded when the triggering event to record the data is initiated during the transmission. It is also contemplated by the present invention that various components of the invention could be combined to reduce the number of components.

[0024] While the invention has been particularly shown and described with reference to the preferred embodiments thereof, it will be understood by those skilled in the art that various changes in form and detail may be made therein without departing from the spirit and scope of the invention, as set forth in the following claims.